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Efficiency Investigation of Subwoofer Driven Around Resonance Frequency

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ABSTRACT

The need for efficient portable speaker systems has increased tremendously over the past 10 years. The batteries, amplifiers and filtering has all seen great improvements in efficiency leaving the speakers units as the most inefficient part of the system, mainly due to the large amounts of current drawn that ends up being dissipated as heat in the voice coil.

This paper will look at how you can design a speaker system to take advantage of the resonance of a speaker unit, since that is where the unit is most efficient and draws the least current. A subwoofer speaker system will be designed with focus on only driving the speaker units near their resonance frequency.

The tests found that with modern DSP it was rather simple to design a speaker system that operate in a very narrow frequency band around the speaker units' resonance frequencies, which in turn ensured a very small current draw. This greatest drawback of this method is the increase in components needed, which drives up cost and complexity.

1 Introduction

Portable speaker system sales continue to increase [1], and so demand for efficient speaker systems that can last longer on a single battery charge increases.

Looking at the individual parts of a portable speaker system it can most often be summed up into the battery, the amplifier, filtering and the speaker(s).
Batteries generally increase in size and weight, whenever you want to increase the amount of energy storage. To ensure portability the battery size and weight must be kept down. So in order for the system to be able to play music for longer periods of time the rest of the system must be optimized.
The development of class D amplifiers has made it possible to design amplifiers with an efficiency of up to 90-95% [2, 3, 4], which makes them near perfect for use in portable speaker system. Moreover they provide great audio performance with very low distortion [5, 6].
The filtering of the signal going from the amplifier to the speaker units is often handled by a digital signal processing unit (DSP), which means a small physical footprint, fine tuning of the frequency response and a low power consumption [7].
This leaves the speaker unit, which is incredibly inefficient compared to the other parts of the system. Most speakers have an efficiency of 0.5-2% [8, 9], meaning that only 0.5-2% of the power send into the speaker unit's voice coil (VC) will generate pressure waves. The rest of the power is converted to heat in the voice coil. Since the nominal impedance of most speaker units is between 2-8 Ω a lot of current is

needed to achieve high sound pressure levels (SPL). This combined with the fact that most portable speaker systems are used outdoors where the sound escapes more easily means that the battery drains fast.

One way to reduce the need for current is to increase the impedance, since

$$P = I^2 \cdot R$$

Looking at an impedance plot for any dynamic speaker unit you will find that the impedance of the VC increase tremendously around the resonance frequency of the driver. This behaviour is what will be used to design a speaker system that draws less current

This paper will investigate the possibility of designing a speaker system where each speaker unit only operates near its resonance frequency to take advantage of the higher impedance.
A speaker system consisting of two different speaker units together covering the frequency band from 65-110 Hz will be designed and assembled. The reason for the frequency band is because of how much content most modern music contains in that frequency band and because it is in the lower frequencies the most current is needed to produce a satisfactory SPL. The SPL at 1W@1m when driven with pink noise will be measured along with the current and voltage needed to produce said SPL. As real music is dynamic pink noise is used because it more closely resembles the energy distribution of music compared to white noise [10, 11, 12, 13].

2 Methods

2.1 Design Choices

2.1.1 Speaker Units

The two units used in this project are a 2 inch sub woofer unit and a 3 inch sub woofer unit, both off the shelf speaker units. The main reason for choosing these two unit was the very high impedance rise near the units' resonance frequencies, which are 75 Hz and 55 Hz respectively. This makes the drivers very suitable for the purpose of this paper.
The 2 inch driver is a Tang Band W2-2040S with a peak impedance of 70 Ω as seen in 1. The sensitivity of this speaker unit approximately 74 dB, which means that it will output a SPL of 74 dB at 1 m with 1 W of input power.

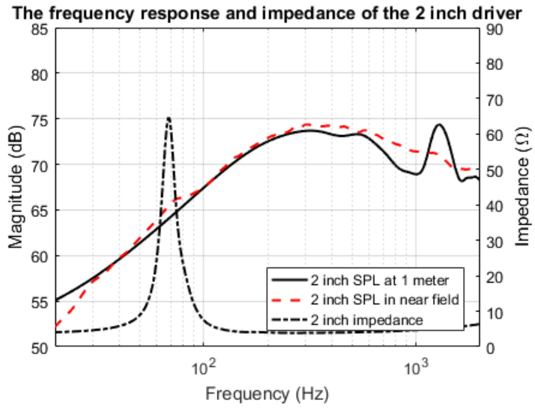


Fig. 1: Frequency response and impedance of the W2-2040S

The 3 inch driver is a Tang Band W3-1876S with a peak impedance of about 25 Ω as seen in 2. The sensitivity of this speaker unit is approximately 77 dB.

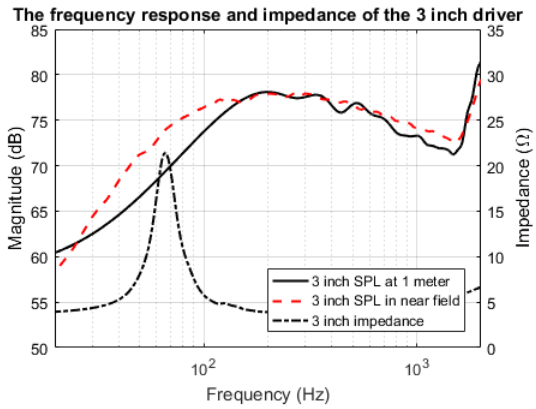
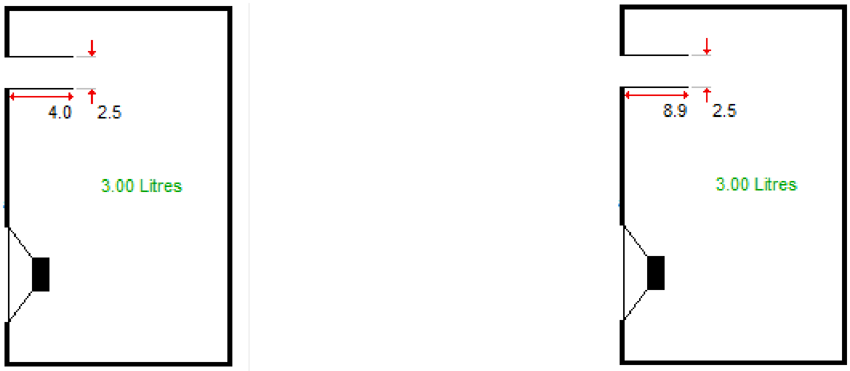


Fig. 2: Frequency response and impedance of the W3-1876S

The sensitivity of both speaker units is rather low compared to most regular high fidelity (Hi-Fi) speaker units that usually have a sensitivity of 83-89 dB [14]. This does not say as much about the final design however, as we will find the total SPL of the two speakers playing together, which will increase the sensitivity. Placing the speaker units in cabinets will also increase the sensitivity, and so the choice of speaker cabinets will help ameliorate this [15].



(a) Cabinet dimensions for the 2 inch speaker unit **(b)** Cabinet dimensions for the 3 inch speaker unit

Fig. 3: Dimension of the two cabinets simulated in FINEBox

2.1.2 Cabinets

When designing the cabinet certain requirements had to be met, since this project looks at efficiency improvements in portable set-ups. The speaker units are sub woofer units intended for reproducing the lower frequencies, and so it was decided that the frequency response of speaker set-up had to cover 65-100 Hz at an SPL of at least 85 dB at 1W/1m in order to keep the system competitive with commercially available speaker system. This had to be achieved while keeping the total internal volume under 6 L, so that the setup was still somewhat portable.

With the previously stated requirements in mind and looking at the speaker units' specifications, especially the total Q factor, Q_{ts} (0.26 and 0.43 for 2 inch and 3 inch speaker units respectively), it was decided that a vented box design would be the most suitable for the job [8, 16]. By using a vented box, we could achieve a higher SPL in the same frequency band as the vent allows for a higher SPL at the lower frequencies. Using FINEBox from Loudsoft the internal volumes of the two cabinets were found along with the dimension of the vents, which can be seen in figure 3. To achieve a high SPL in the frequency band specified we took advantage of the output of the vent, which is why you observe such a peaky behavior in figure 4. This will be used, when designing the crossover. Also, the 3 inch is about 3 dB lower in SPL, but this will be boosted via DSP. See section 2.1.3.

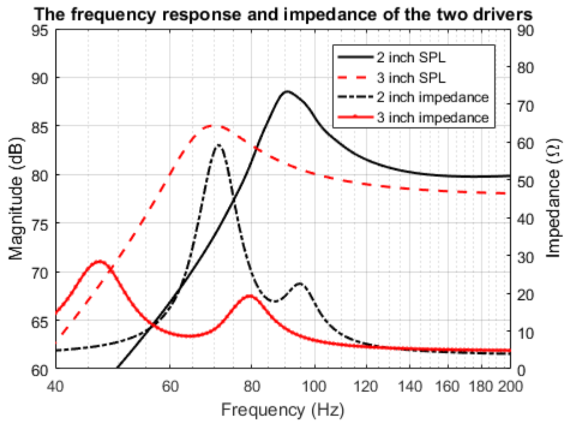


Fig. 4: Frequency response and impedance of the two speaker units in FINEBox

The frequency response and impedance of the two speaker units in their individual cabinet were simulated in FINEBox and can be seen in figure 4.
It is seen in figure 4 that the SPL of the 2 inch speaker unit is 3 dB higher than that of the 3 inch speaker unit. This is a consequence of having the 3 inch speaker reproduce lower frequencies with the same cabinet volume. To achieve a more even combined frequency response we can either attenuate the output of the 2 inch or boost the output of the 3 inch. Attenuating is not desirable, since we would lose max SPL. Looking at the excursion of the VC of the 3 inch in figure 5 we can see that diaphragm is hardly moving in the frequency band we wish to drive it in. Looking at the specifications of the 3 inch speaker unit we can see that Xmax (maximum excursion of the VC) is 5 mm, which

means that we can boost the output of the 3 inch via DSP (Digital Signal Processing) without risking the VC overextending.

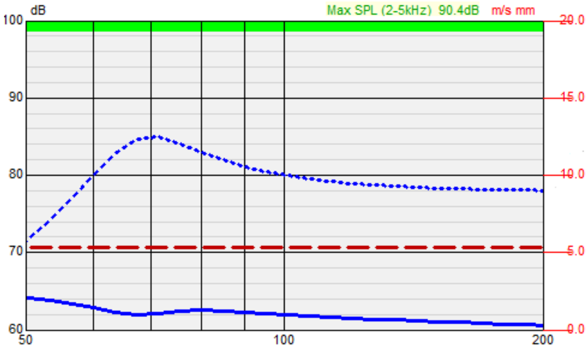


Fig. 5: Excursion of the VC (solid line) versus the frequency response (dashed line) of the 3 inch speaker unit

2.1.3 Crossover

The crossover between the two speaker units was chosen so that they would operate ±3 dB around their highest SPL output. This means that the 3 inch speaker unit has to operate between 65-85 Hz and the 2 inch speaker unit has to operate from 85-110 Hz, see figure 4. Since the two speaker units operate in such narrow bands and so close to each other that very steep slopes are needed, i.e. -48dB/oct. It was quickly decided against passive components as it would involve too many components that would take up too much space, introduce large losses while being more difficult to achieve.
The next possibility was active components, i.e. Sallen-Key filters. Active filters was also decided against again due to filter complexity, when going for such a high slope.
The final option was to use DSP. This would allow for the very steep slopes required along with easier fine tuning of the frequency response by using parametric equalization. The DSP chosen was a miniDSP [17], which is a DSP package that handles analog-to-digital conversion, performs the crossovers and equalization and then performs digital-to-analog conversion. Using the GUI the crossovers were quickly setup with the required slopes and frequency bands.

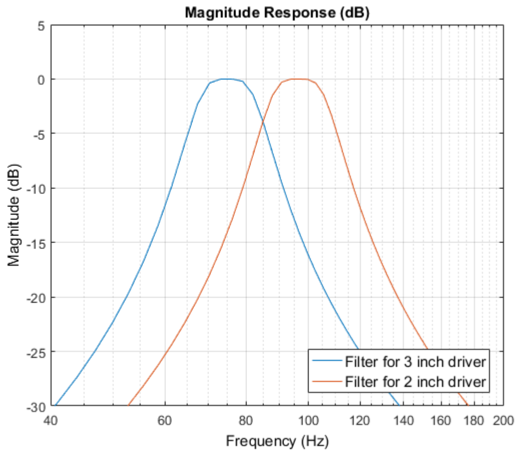


Fig. 6: The crossover of the two speaker units implemented via a miniDSP

2.2 Measuring Setup

Fig. 7: The two speakers as measured

2.2.1 Frequency response

The frequency response is measured in an anechoic chamber at Technical University of Denmark. The room is approx. 60 m^3 and with a limiting lower frequency of about 100 Hz. This makes the room less than perfect for measuring the lower frequencies; however, the distances to the walls, celling and floor are more than 1.5 times longer than to the microphone, and while the room does not eliminate the reflection below 100 Hz completely they are still attenuated compared to the direct wave, making the room usable for measuring the frequency response. The frequency response is measured by performing a sweep from 1-200 Hz using Room Equalization Wizard (REW) along with a Umik-1 microphone from miniDSP.

2.2.2 Power Draw

To measure the current, voltage and power drawn by the individual speaker a Teledyne Lecroy MSO 104MXs-B is used. The current running into the speaker unit is measured using a AP015 current probe griping the positive wire attached to the positive terminal of the speaker unit. The voltage is measured using a differential voltage probe attached to the positive and negative wires connected to the positive and negative terminals of the speaker unit respectively. The total power drawn by each of the speaker units is found by $P = V \cdot I$.
The input signal used when measuring the power draw is pink noise, since it contains an equal amount of energy in each octave band. This gives a better representation of how music is perceived by the human ear compared to using white noise, which is why it is often used to fine tune speaker systems [12].

2.2.3 Sound Pressure Level

The SPL is measured in the same room as the frequency response using REW and the Umik-1 with pink noise as the input signal. The microphone is placed 1 meter away from the speaker system and amplifier outputs one watt into each of the speaker units.
The SPL is measured using pink noise. The SPL is measured over 30 seconds to ensure that the measured SPL is not a local low or high due to the randomness of the input signal. The measured SPL is then averaged over the period to obtain the average SPL output of the speaker system.

3 Results

3.1 Frequency Response

Figure 8 shows the frequency response of the speaker system, which has the desired output from 65-110 Hz with all other frequencies attenuated heavily. The SPL matches the simulated SPL of the two speaker units seen in figure 4 somewhat. The simulated responses are without crossovers applied, hence output shown at higher frequencies, whereas the measured response shows the total frequency response of the two speakers combined with crossover applied.

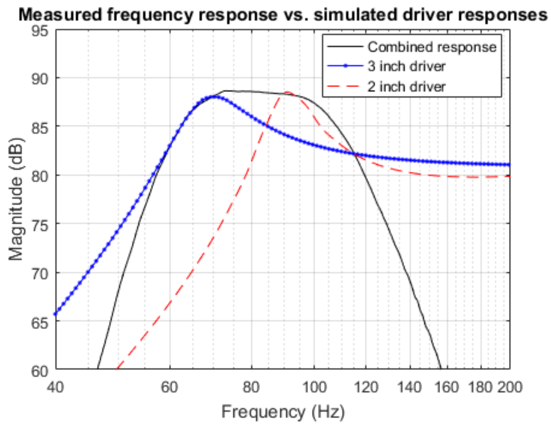


Fig. 8: Total frequency response of the speaker system

3.2 Sound Pressure Level

Figure 9 shows the SPL of the speaker system, if the speaker system was driven with 1 W. Comparing the SPL while playing pink noise with the frequency response in figure 8, which is normalized to 1W@1m, we see a good correlation and such they help to validate each other.

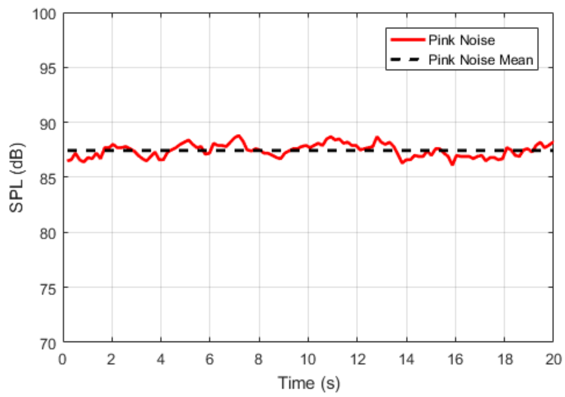


Fig. 9: The SPL of the speaker system while playing pink noise at 1W

What is most interesting and the main subject of this paper is the current drawn by the speaker units while playing. Figure 10 shows the current drawn by the speaker system and the individual speaker units. These measurements were done when the amplifier was outputting 0.01 W into the speaker system. The reason for the low output is due to an unexplainable attenuation of the signal in the miniDSP when using pink noise. The behavior was not observed when using sine waves or music. What is most interesting about this plot is the difference in current drawn by the 2 inch speaker unit and the 3 inch speaker unit. The 2 inch draws quite a bit less current than the three inch with the units drawing on average 13.1 mA and 21.7 mA respectively. Looking at the impedance plots in 4 we see that the impedance of the 2 inch is substantially greater in the 2 inch's passband (85-110 Hz) compared to the 3 inch's impedance in its passband (65-85 Hz), which explains the need for less current. This is while both drivers are outputting the same SPL at the same voltage, helping to confirm that speaker units with a high impedance can be relevant when designing portable speaker systems.

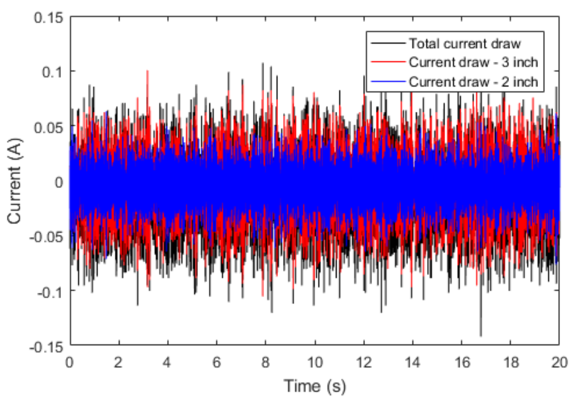


Fig. 10: The current drawn by the speaker system while playing pink noise at 10 mW

4 Discussion

It is fairly clear from the results seen in figure 10 that there are advantages to driving the speaker near its resonance frequency. Due to the great increase in the speaker unit's impedance the current required to drive the speaker unit is greatly reduced. At the same time the speaker unit does not have to work very hard as it is primarily the port that provides the SPL, which means that less VC excursion is required, see figure 5 and thus we can push the speaker unit to produce even higher SPL without VC leaving the motor.

However, it is not all good as there are some disadvantages, especially if the speaker unit is to be used in a portable setup. Because the impedance increases so much at the resonance, up to 34 Ω, the voltage required to drive the unit also increases greatly. This can pose a problem, since most batteries are either 12 V or 24 V. This problem can be bypassed somewhat by using boost converters and other circuitry to increase the voltage. Another problem is the narrow band at which the unit operates most efficiently. If you want the speaker system to cover the lower frequencies from, say, 50-150 Hz, since that is where the most current is usually spent, you will need at least 4+ speaker unit similar to the speaker units used in this paper, if you want them to operate in the narrow frequency band where they are the most efficient. This adds cost, complexity and size, which are all negatives when building portable setups. However, if custom speaker units were to be designed for a system such as the one used in this paper, then it might be possible to reduce the negative effects. Since the speaker units only have to cover a narrow frequency band less time is needed for optimizing the frequency response of the speaker units as you only have to focus on the narrow frequency band the speaker units need to cover. Also, the requirements for expensive materials that behave well across a wide frequency band diminishes, which can help to reduce the costs and complexity further.

The results also raise the question of why speaker units always aim for nominal impedances of 2-8 Ω, when there are clear advantages to higher impedance speaker units, especially for portable systems. Headphones have generally been designed with impedances ranging from 32-600 Ω, which makes you wonder even more why regular speaker units have to be designed with such a low impedance.

Looking at the benefits we see that we can reduce the amount of current needed to drive the speaker unit considerably, which prolongs battery life and it can have the added benefit of less distortion as well [18]. Again, this will still require the amplifier to be able to raise the voltage considerably in order to be able to drive the speaker system to a satisfactory SPL.

5 Summary

It is clear that building a speaker system that utilizes the efficiency of the speaker units when driven near the resonance does have some merit, especially when you are concerned about battery life. It is key to acknowledge that battery life is the main motivational factor for designing such a speaker system however, since the negative sides can easily outweigh the benefits. Because we have to use several drivers to cover any appreciable frequency band the size, complexity and price of the design increases substantially and so it might quickly loose its appeal for smaller portable speaker systems. Outdoor, battery-driven portable systems on the other hand might not be as restricted in regards to the size of the system, and so a setup similar to the one described in this paper with multiple speaker units might be an attractive option.

If you opt for a design such as the one investigated in this paper, there are certain factors that should be considered when designing the speaker units. First up would be to design the speaker unit with a high impedance; however, whether this should be across the full frequency band like in a headphone or around the resonance frequency depends on how the speaker is thought to be used. Driving the unit only at resonance has the advantage of voice coil needing to move very little, but also means that the usable frequency band is quite narrow and increases the need for high order filters. Choosing to go with a high impedance across the frequency band increases the usable frequency band and so reduces the number of speaker units required and might lower the crossover slopes needed. Next thing will be to design the unit, so that it performs the best in the cabinet design chosen. Whether a closed box or a box with a vent/passive radiator is used is up to the individual as they both have advantages and disadvantages. Lastly, the designer has to ensure that the amplifier can handle the high impedance, i.e. that it can output a high enough voltage to produce an appreciable SPL.

This paper has proven that a speaker system designed around the resonance frequency of the speaker units does have its advantages, especially in portable setups where current draw is a consideration. However, it comes with some caveats that need to be overcome. So it is up to the designer to weigh the pros and cons, when choosing which direction to go.

6 Future Work

A comparison with a more traditional design, i.e. a single driver subwoofer, needs to be done to see how big a reduction of the current draw one can achieve. For such a comparison there are certain requirement however. The design will have to cover the same frequency band with an SPL equal to that of the design used in this paper as a minimum. For it to be a proper comparison the cabinet volume cannot exceed the total volume of the design described in this paper, which is 6 L.
It would also be interesting to have a speaker unit designed with a overall higher impedance to allow for a single driver design. This would both reduce complexity of the entire setup, while benefiting from a reduced current draw.

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